Matlab Project - Adaptive Equalization

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Channel Equalization

In a digital communication system, we want to avoid intersymbol interference (ISI) to have reliable communication. However, ISI can be caused by channel distortion. A common type of distortion is channel dispersion as a result of the nonlinear phase characteristics of the channel. A model for the received sequence $y[n]$ is

$$y[n] = \sum_{k=-\infty}^{n} x[k]h[n,k] + w[n] \tag{1}$$

where $x[k]$ is the transmitted sequence, $h[n,k]$ is the time-varying impulse response of the channel, and $w[n]$ is zero-mean white Gaussian noise. In order to reduce the chance of making an error, the receiver will often employ an equalizer to reduce the effects of channel distortion. Since the channel characteristics are unknown and possibly time-varying, the equalizer is typically an adaptive filter.

In this exercise, a message sequence is coded, modulated and sent through a dispersive channel. First, each ASCII character is turned into a 5-bit codeword. The codebook is shown in the table below.

character	ASCII decimal value	binary string code
space	32	00000
'a'-'z'	97 -122	00001-11011

Then the bit stream is BPSK modulated before transmission, i.e., $x[n] = -(-1)^{d_n}$, where $\{d_n\}$ is the bit stream. The received signal can be expressed as

$$y[n] = x[n] + a_1[n]x[n-1] + a_2[n]x[n-2] + w[n] \tag{2}$$

where $a_1[n], a_2[n]$ are channel characteristics and time-varying and $w[n]$ is the AGWN noise. You are given a sequence of the received signal $y[n]$ and the first 200 alphabet symbols sent as training data. You are asked to design a channel equalizer so that the ISI can be reduced as much as possible. Your final result should be a channel equalizer and the decoded message.

Hint

From (2), we can write

$$x[n] = y[n] - a_1[n]x[n-1] - a_2[n]x[n-2] \tag{3}$$

The LMS update equation then can be written based on this equation.
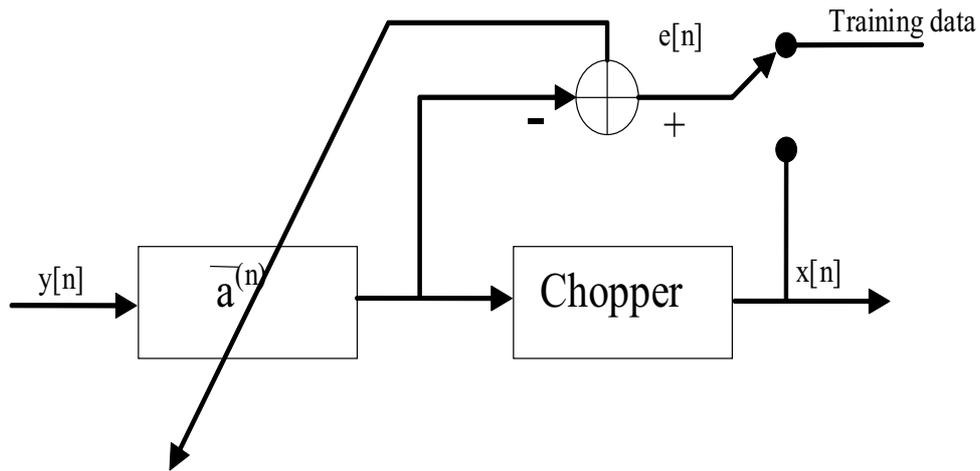
First use the training data to update the coefficients a_1 and a_2, then proceed to the message while still updating these coefficients. The LMS update then can be written as follows:

$$x[n] = y[n] - \bar{a}_n^H \bar{x}[n] \text{ (filtering)}$$
$$e[n] = d[n] - x[n] \text{ (assessing)}$$
$$\bar{a}_{n+1} = \bar{a}_n + \mu \bar{x}_n e_n^* \text{ (updating)}$$

where

$$\bar{x}[n] = \left[\begin{array}{c} x[n-1] \\ x[n-2] \end{array} \right]$$

Reference

[1] ECE551 - Advanced Digital Signal Processing Homework, University of Illinois at Urbana-Champaign, Fall 2005.